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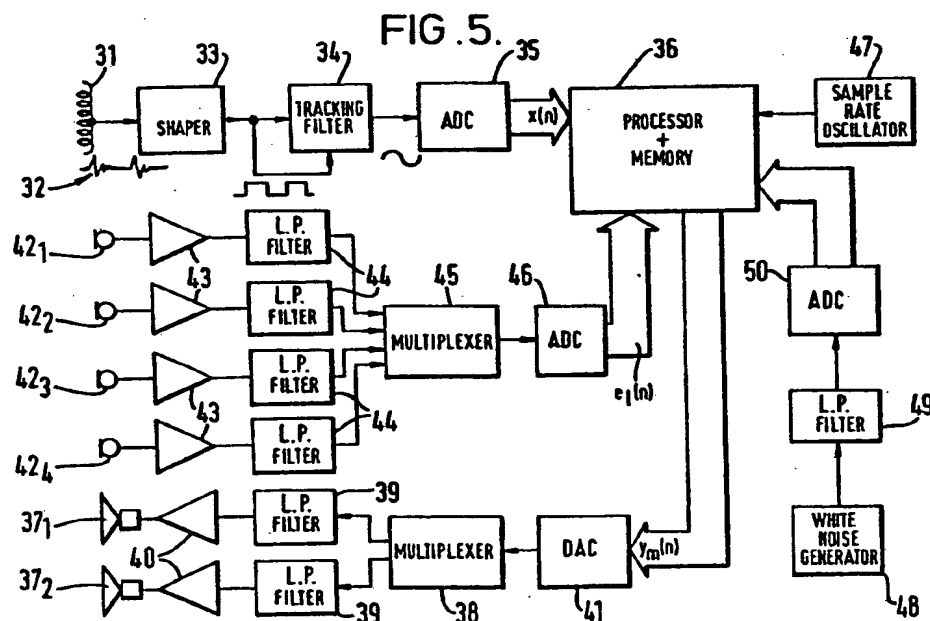
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(54) Active sound control apparatus

(57) To reduce noise inside a motor car passenger compartment, two loudspeakers 37₁, 37₂ are driven by signals derived from a reference signal $x(n)$ by adaptive filtering carried out by a programmed microprocessor and memory unit 36 which adapts the filtering in dependence on error signals $e_i(n)$ from four microphones 42₁, 42₂, 42₃ and 42₄ distributed in the passenger compartment. Reference filtering coefficients are initially determined by analysis of finite impulse responses when white noise is acoustically coupled from the loudspeakers 37 to the microphones 42, a white noise generator 48 being coupled to the unit 36. The reference signal $x(n)$ is restricted to one or more selected harmonics or subharmonics of the fundamental noise frequency by a filter 34 which tracks the selected frequency. The selected frequency may be obtained from a coil 31 in the ignition circuit.



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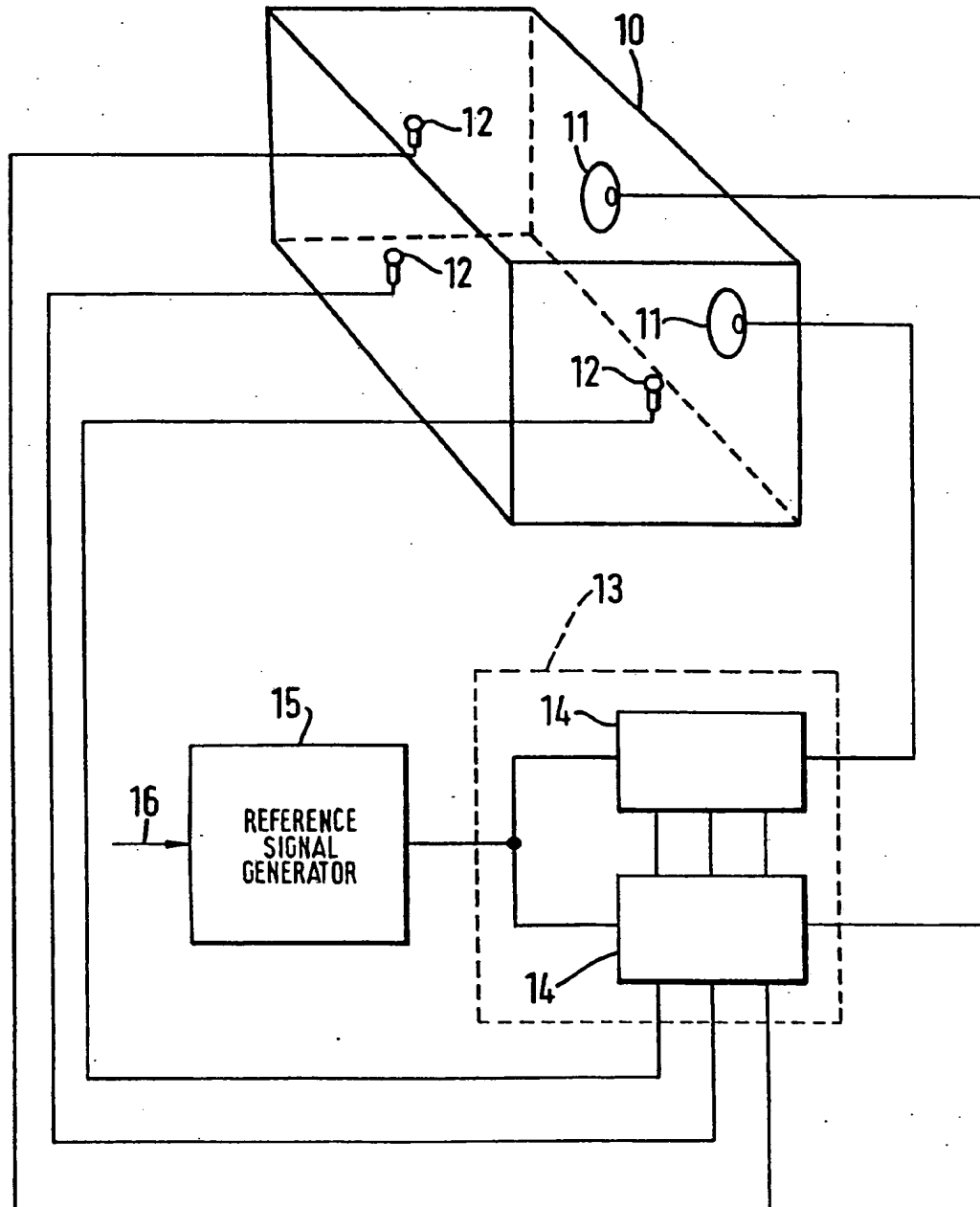
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FIG. 1.

FIG. 2.

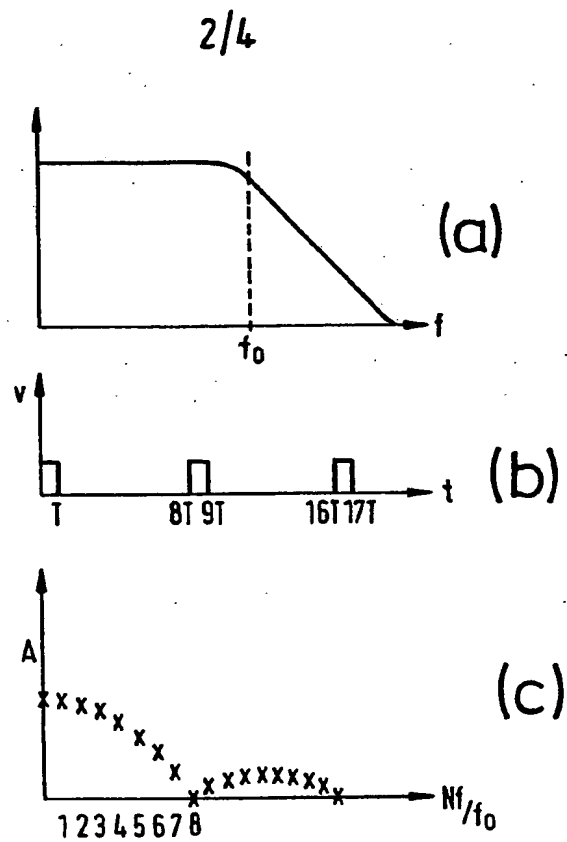


FIG. 3.

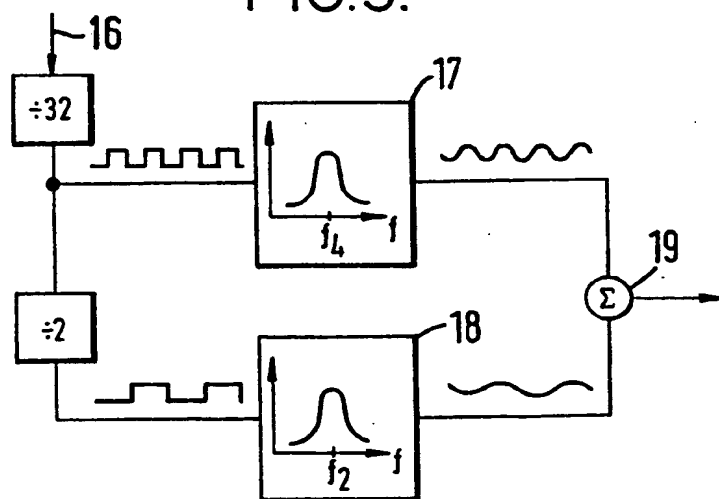
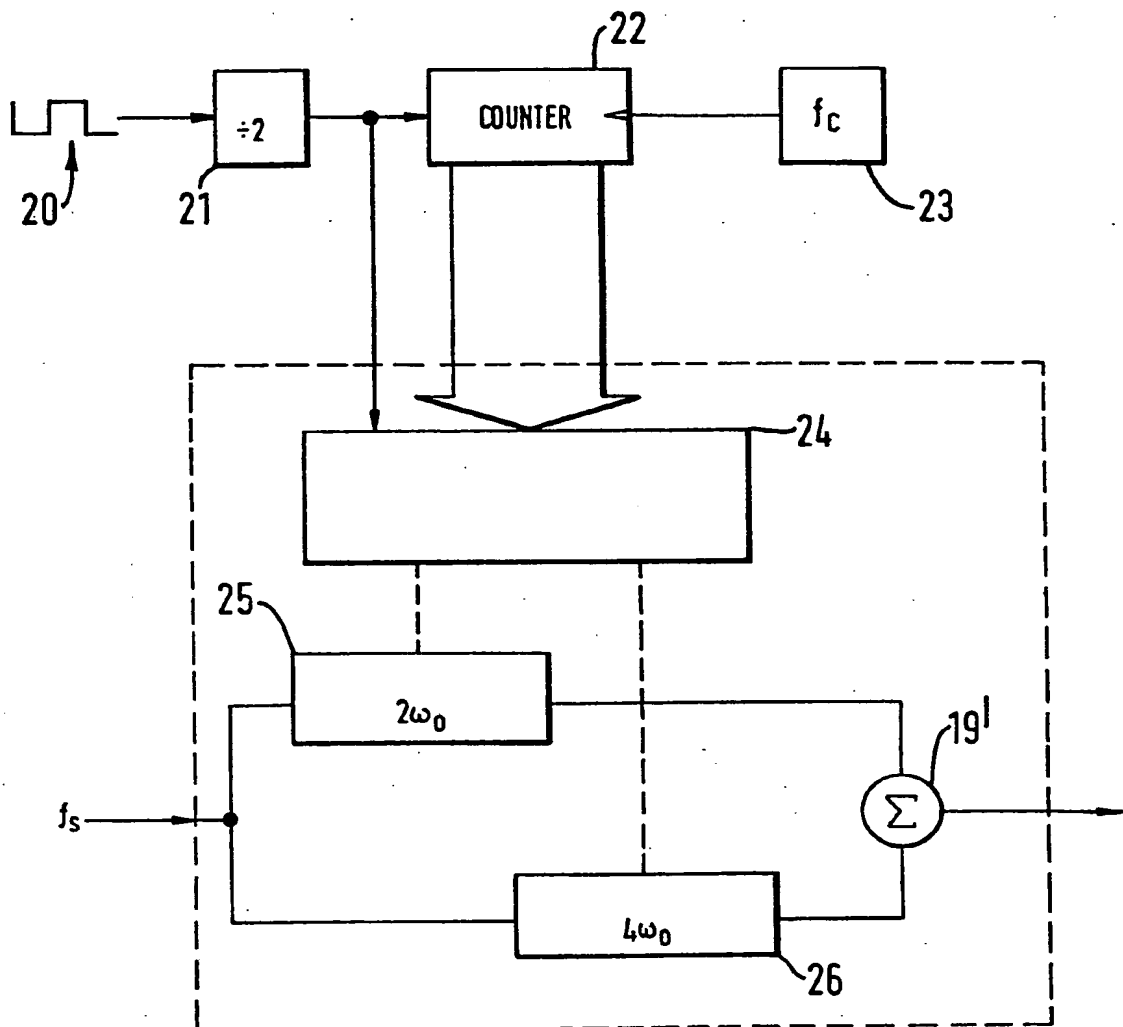
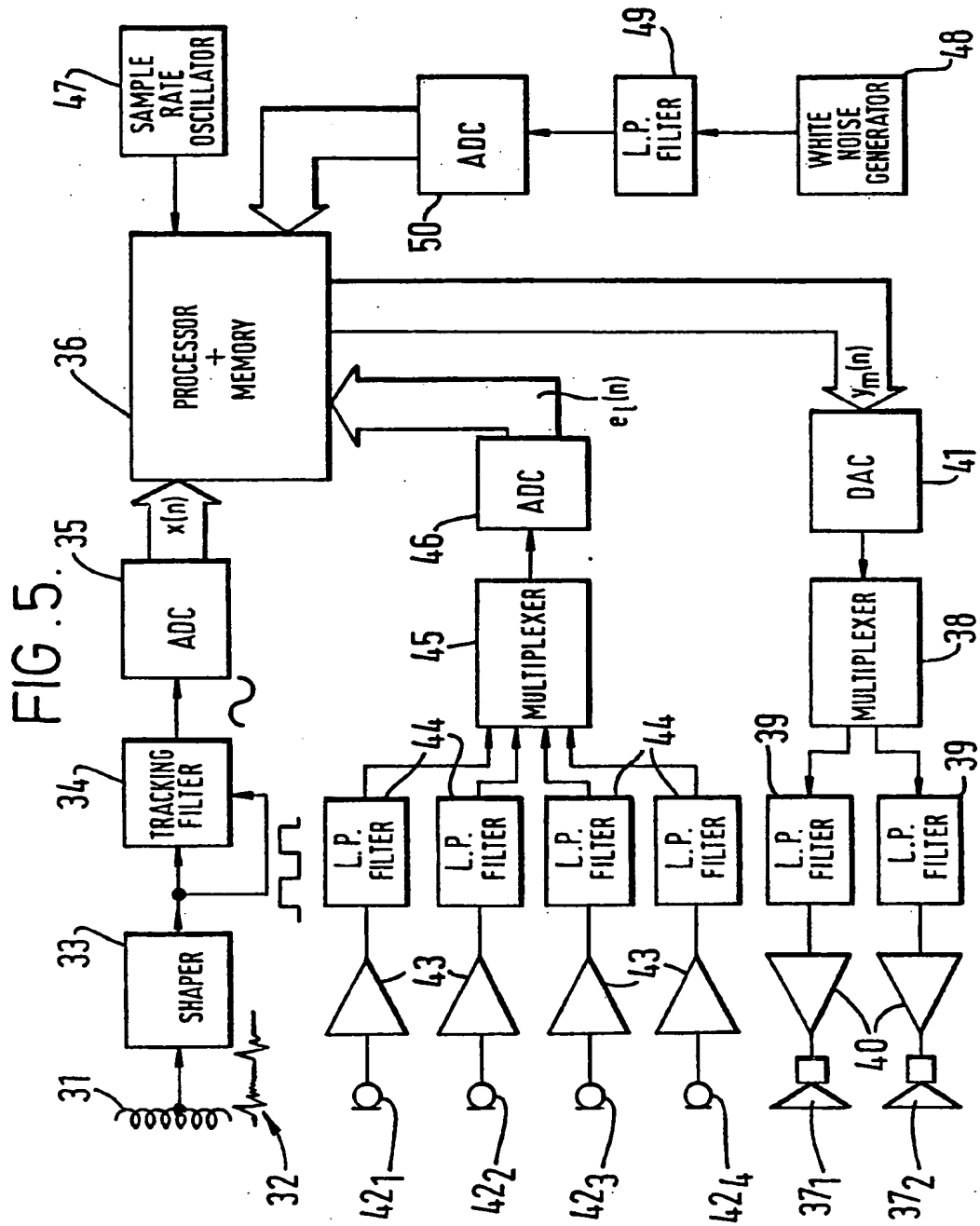


FIG. 4.





6K28042ACTIVE NOISE CONTROL

The invention relates to active noise reduction, particularly in passenger compartments of cars, where a significant noise component is harmonically related to the rotation frequency of a reciprocating engine. The sound levels at low frequencies in such enclosures are difficult to attenuate using conventional passive methods and can give rise to subjectively annoying "boom". A method of actively attenuating a simple sound field by introducing a single secondary sound source driven so that its output is in antiphase with the original ambient noise is described in general terms by B. Chaplin in "The Chartered Mechanical Engineer" of January 1983, at pages 41 to 47. Other discussions are to be found in an article entitled "Active Attenuation of Noise-The State of the Art" by Glenn E. Warnaka at pages 100 to 110 in Noise Control Engineering, May-June 1982 and in Internoise 83 Proceedings, pages 457 to 458 and 461 to 464, and Internoise 84 Proceedings, pages 483 to 488. Particular methods and apparatus are also described in British specifications nos. 1 577 322 and 2 149 614.

The present invention is concerned particularly with an active noise reduction system which can control the sound throughout an enclosure, or at one or a number of 'quiet zones' within it, and which can quickly adapt to changes in the excitation of the sound field due to changes in engine condition caused, for example, by load or speed variations. In order to ensure that the sound produced by the secondary sources is of the same frequency as that produced by the engine, a signal related to the rotation rate, for example that used in the ignitions system, is used to generate a reference signal containing a number of sinusoids at harmonics or subharmonics of the basic engine rotation frequency. These are known as engine order frequencies. These sinusoids may be obtained using a variety of methods outlined below. Rather than attempt to control all harmonics, one feature of the present invention is only to generate a selective set of engine order frequencies in the

engine order in a four cylinder car) and its second harmonic (fourth engine order) are used if the spectrum of the sound in the car is dominated by these components. Alternatively, a signal containing all the engine order frequencies may be fed to a band pass filter which isolates only the particular frequency or frequencies exciting a particular resonance in the interior which could cause a "boom" to be excited. The advantage of reducing the number of frequencies fed to the controller is that an adaptive filter having fewer coefficients than would otherwise be the case can be used. This makes implementation more efficient and allows faster automotive adaption time. The faster adaption time is particularly important in applications in which the active control system has to adapt sufficiently quickly to track changes in engine speed which may occur on a very short timescale.

According to the present invention there is provided an active noise control system for reducing sound or vibration in an enclosed space, generated by an exterior source, in which a reference signal, containing selected harmonics or subharmonics of the sound or vibration generated by the external source, is used to feed a controller driving a number of secondary sources distributed in the enclosed space, such that the sound or vibration energy detected by sensor means distributed in the enclosed space is reduced.

The secondary sources may be loudspeakers used as the low frequency drives of a car audio system, the enclosed space being a motor vehicle interior.

Examples of methods in accordance with the invention for generating the reference signals from the engine signal are now discussed.

1. Selection of harmonics by filtering.

A signal is obtained from the primary source which contains components at all harmonics and subharmonics which are prevalent in the sound in the car. This is filtered so as only to leave the most important or dominant harmonics or subharmonics. This filtering is carried out with a filter whose centre frequency can be controlled by an external signal in such a manner that the critical filter frequencies have a constant ratio compared with the engine rotation rate. This can be achieved by using, for example, charge coupled

devices whose switching frequency is locked to the rotation frequency, but can also be implemented as a program running on a microprocessor, as described hereinafter.

2. Selection of harmonics or subharmonics within a band by fixed filtering.

A signal rich in harmonics and subharmonics is filtered by a band pass filter, having a centre frequency fixed at that of a pronounced "boom" in the car enclosure and a characteristic such that the reference signal only contains the harmonic(s) or subharmonic(s) which are particularly exciting the boom. This may be extended such that the filter contains a number of resonances at a number of boom frequencies of the car, or even such that the filter has a frequency response which models the acoustic response of the car interior to the primary excitation. The input signal to the filter may be a signal from the engine containing all important harmonics and subharmonics and may be in the form of a pulse train.

3. Generation of specific harmonics or subharmonics locked to the engine rotation frequency.

This method may be accomplished by using phase lock loops to generate sinusoidal signals, with frequencies bearing an integer relationship to a square wave signal from the engine, which are then added together to form the reference signal. Alternatively the signal from the engine can be used to control a number of tunable oscillators, each producing a sinusoid at a selected harmonic or subharmonic frequency. In one arrangement for such a bank of tunable oscillators, the period of the square wave signal at the engine rotation rate is measured with a counter passed to a microprocessor which implements a number of digital oscillators using difference equations of the form

$$x_I(n) = \delta(n) + 2 \cos(I\omega_0)x_I(n-1) - x_I(n-2)$$

where $\omega_0 = 2\pi f_c / Nf_s$, I is the order of the harmonic or subharmonic to be generated, f_c is the frequency of the counter, which counts N pulses during a period, and f_s is the sample rate used for the difference equation. $\delta(n)$ is a unit sample sequence to initiate the oscillators formed by the difference equations above. The second and fourth harmonic may be generated for example. The calculations of the coefficients used in the difference equations forming the

digital oscillators, together with the difference equations themselves may be implemented on a dedicated processor, or may form part of the program which also implements a controller that generates the outputs used to drive the secondary sources from the reference signals described above. The controller is designed to be adaptive so as to quickly track changes in engine speed and load. The outputs of the secondary sources are adaptively controlled so that some measurable cost function is minimised. This cost function would typically be the sum of the mean square outputs from a number of microphones in the enclosed space. The controller can be implemented as a digital adaptive FIR filter, using the basic update algorithm described by S. Elliott and P. Nelson in "Electronics Letters", at pp. 979-981, 1985. A number of additions must be made to this basic algorithm, however, to enable it to work quickly and efficiently in this particular example. The basic algorithm is presented below, in order to highlight the necessary alterations. If the i 'th coefficient of the adaptive filter driving the m 'th secondary source at sample number n is $w_{mi}(n)$, then each of these coefficients should be adjusted at every sample according to the equation

$$w_{mi}(n+1) = w_{mi}(n) + \alpha \sum_{l=1}^L e_l(n) r_{lm}(n-1)$$

where α is a convergence coefficient, $e_l(n)$ is the sampled output from the l 'th sensor and $r_{lm}(n)$ is a sequence formed by filtering the reference signal discussed above ($x(n)$, say) with a digital filter which models the response of the l 'th sensor to excitation of the m 'th secondary source. These digital filters, generating each $r_{lm}(n)$, have only two coefficients in the implementation described in the 'Electronic Letters' article, since control at only a single, fixed, frequency was being attempted. In the present example, however, these filters must model the relevant response over a range of frequencies, governed by the frequency range which the active system is attempting to control. It has been found that under certain circumstances this filter need only model the overall delay in the response to ensure the stability of the adaptive filter. It is more common, however, to have these filters incorporating a delay and then some reverberant response. This may be implemented using

either digital FIR or IIR filters whose coefficients are adjusted adaptively during an initialisation phase, so as to accurately match the desired responses. It is also possible to continue this initial adaption process during the operation of the active control system by feeding training signals to each secondary source which are suitably uncorrelated with each other and with the primary excitation. This may be necessary to track changes in the acoustic response of the enclosure. Alternatively if the change is due to some well-defined cause, such as a passenger sitting down or a window being opened, this change may be detected with mechanical transducers and the information used to switch between a variety of filters modelling the response of the enclosure under a variety of conditions.

Another important consideration concerning the use of adaptive algorithm in this example concerns the effect of unwanted, low level, harmonic or subharmonic components in the reference signal. Suppose that the method used to generate the reference signal, as described above, is designed to produce only I frequencies. Even though there may only be I desired harmonics in such a system, there will in practice also be a number of other harmonics or subharmonic frequencies at low level, because of the finite cut-off rate of the filters, for example. These components may also be being generated by the primary source and therefore present in the enclosure and hence at the outputs of sensors such as microphones and thus the adaptive algorithm will attempt to cancel them by enormously amplifying the low level, spurious harmonic reference signals. This can cause numerical overflow problems in the adaptive filter coefficients. This may be prevented in a number of ways:

- (1) The use of only $2.I$ coefficients in the adaptive filter.
- (2) The deliberate injection of random noise into the reference signal.
- (3) The insertion of a 'leak' into the algorithm so that, in the equation above, the past coefficient value, $w_{mi}(n)$, is multiplied by a factor close to but not equal to unity before being updated.
- (4) The addition of an extra term in the update equation which minimises a cost function involving 'effort' as well as 'error', as

described in ISVR Technical Report No. 136, 1985, published by the Institute of Sound and Vibration Research at the University of Southampton.

The invention will now be described by way of example with reference to the accompanying drawings, in which:-

Fig. 1 is a block schematic diagram of an active sound control apparatus for an enclosed space,

Fig. 2(a), (b) and (c) are graphical representations of the behaviour of an element of an apparatus embodying the invention,

Fig. 3 is a block diagram of a reference signal generator for an apparatus embodying the invention,

Fig. 4 is a block diagram of another reference signal generator for an apparatus embodying the invention, and

Fig. 5 is a block diagram of a microprocessor-based apparatus embodying the invention.

In Fig. 1 an enclosure 10, which may be the interior of the passenger or driver compartment of an internal combustion engine driven vehicle, for example a motor car, is represented schematically together with an active sound control apparatus embodying the invention and having in this example two secondary sound sources 11, which may be two low frequency loudspeakers of a car stereo audio system, and three microphones 12. The secondary sources 11 are driven by a controller circuit 13 which comprises a plurality of adaptive filters 14. Each adaptive filter 14 drives a respective one of the secondary sources 11 with an output signal which the filter 14 produces as a result of its action on a reference signal supplied thereto by a reference signal generator 15. The reference signal is generated by the generator 15 from an input signal 16 which is periodic at the rotation rate or the firing rate of the internal combustion engine (not shown).

The purpose of the output from the secondary sources 11 driven by the controller 13 is to reduce the sound level, due to the internal combustion engine, within the enclosure 10. Since the primary source of the sound to be reduced is periodic, the reference signal generated by the generator 15 is, in accordance with the invention, arranged to contain one or more sinusoidal components at harmonics or subharmonics of the rotation or firing rate of the

engine, and the adaptive filters are adjusted automatically by signals from the microphones 12, the adjustment being such as to minimise a cost function such as the sum of the mean square outputs of the microphones 12.

The secondary sources 11 may in another embodiment be shakers, and the microphones 12 replaced by accelerometers.

It will be noted that the secondary sources 11 and the microphones 12 or other sensors are distributed in the enclosed space 10.

Since the reference signal contains harmonics or subharmonics of the input signal 16, which is periodic at the engine rotation or firing rate, the reference signal contains engine order frequencies. The generator 15 is arranged to select engine order frequencies that ensure that the sound produced by the loudspeakers 11 is of the same frequency or frequencies as the sound produced in the enclosure 10 by the engine, even during changes in engine conditions such as load or speed. The number of engine order frequencies in the reference signals is restricted so that the adaptive filters have a relatively small number of coefficients and can therefore adapt quickly.

The input signal 16 can be obtained from a moving part of the engine or part of the ignition circuitry; for example.

Fig. 2(a) illustrates graphically the response of the reference signal generator 15 when implemented in the form of a tracking filter, i.e., a filter having a centre frequency so controlled that the output frequencies have a constant ratio to the dominant input frequencies, so that in Fig. 2(a), the frequency f_0 is $N \times$ (engine rotation rate), and the reference signal contains only the first N harmonics of the engine rotation rate, where N is an integer. If the input signal is a voltage pulse train as represented by Fig. 2(b), where $8T$ is the periodic time of the engine rotation rate, the first eight harmonics of the engine rotation rate are present in the reference signal. The spectrum, by Fourier analysis, of the reference signal is then illustrated by Fig. 2(c), in which A is amplitude. With the response of Fig. 2(a), only the first six harmonics would be usable.

The tracking filter can be in the form of charge coupled devices having a switching frequency locked to the engine rotation rate.

An alternative reference signal generator using a plurality of tracking band pass filters is illustrated by Fig. 3 in which the input signal 16, a square wave at, for example, 128 times the engine rotation rate, is divided by 32 and then by 2. Division by 32 produces a square wave at four times the engine rotation rate which is supplied to a first bandpass filter 17 having a centre frequency f_4 which is arranged to track twice the fundamental frequency of the square wave supplied thereto. The further division by 2 produces a square wave at the engine rotation rate, which is supplied to a second bandpass filter 18 having a centre frequency f_2 which is arranged to track the fundamental frequency of the square wave supplied thereto. The bandpass filters 17 and 18 produce respectively sinusoidal output signals at f_2 and f_4 which are linearly summed at an adder 19 to produce the reference signal.

Further dividers and tracking bandpass filters can of course be added to or inserted in the circuit of Fig. 3 so that the reference signal contains the desired set of engine order frequencies.

Another reference signal generator may consist of a fixed frequency filtering circuit which selects harmonics and/or subharmonics from an input signal rich in the harmonics of the engine rotation rate or firing rate. The circuit may be a bandpass filter having a centre frequency fixed at the frequency of a pronounced resonance excited in the enclosure by the engine or other primary source of sound or vibration. For example, a bandpass filter may be arranged to have a response that models the acoustic response of a motor car passenger compartment to the engine.

A further implementation of the reference signal generator may comprise a plurality of phase lock loops used to generate sinusoidal signals having respective frequencies with integer relationships to a square wave input signal from the engine or other primary source, the sinusoidal signals being added together to form the reference signal. Thus a reference signal consisting of specific harmonics and/or harmonics locked to the primary source fundamental, such as engine rotation rate, is generated. An alternative generator for such a reference signal is illustrated in Fig. 4 in which a square wave 20 at the primary source fundamental is used to control a

plurality of tunable oscillators 25,26 each producing a sinusoidal signal at a chosen harmonic or subharmonic frequency to be added at the adder 19 which produce the reference signal by simple addition of the sinusoidal signals.

In the generator of Fig. 4, a square wave 20 at an engine rotation rate is supplied to a bistable circuit 21 to divide the rate by two and thereby produce a pulse train in which the duration of each pulse is equal to the prevailing periodic time of the square wave 20. This periodic time is then measured by a counter 22 which is enabled throughout the duration of each positive pulse from the bistable circuit 21 and counts clock pulses supplied by a clock pulse generator 23 which generates the clock pulses at a fixed, suitably high rate f_c .

The contents of the counter 22 are read at the end of each positive pulse from the bistable circuit 21 by a trigonometric function generator 24 which is triggered by the trailing pulse of each positive pulse from the circuit 21 and generates two digital outputs representing respectively $\cos(2\omega_0)$ and $\cos(4\omega_0)$ where ω_0 is given by

$$\omega_0 = 2\pi f_c / N f_s$$

in which N is the number of clock pulses counted by the counter 22 in the duration of one positive pulse from the circuit 21, and f_s is a sample rate used in two digital oscillators 25 and 26 which receive respectively the digital outputs $\cos(2\omega_0)$ and $\cos(4\omega_0)$ from the function generator 24. The digital sinusoidal outputs from the two oscillators 25 and 26 are superposed by a digital adder 19' which supplies the reference signal as a digital signal. The trigonometric function generator 24, oscillators 25 and 26 and adder 19' can be implemented by a microprocessor with a suitable program. The microprocessor can be used to produce the reference signal in the form.

$$x_I(n) = \delta(n) + 2 \cos(I\omega_0)x_I(n-1) - x_I(n-2)$$

where I is the order of the harmonic or subharmonic being generated, $\delta(n)$ is a unit sample sequence which initiates the simulation of the oscillator, and n is the sample number.

Fig. 5 represents in block form an active sound control apparatus embodying the invention for reducing the level of engine

generated sound in the passenger compartment of a motor car having an ignition circuit including a low tension coil 31 from which a voltage signal 32 at the firing rate of the engine is taken and supplied to a waveform shaper 33 which in response thereto produces a pulse train at the engine firing rate. It is assumed in the present example that the engine firing rate is twice the engine rotation rate f_0 . Thus the shaper 33 provides a signal having a fundamental frequency which is a single harmonic, $(2f_0)$, of the engine rotation rate. A reference signal generator is provided in the form of a proprietary tracking filter 34, manufactured by Bruel and Kjaer under type number 1623, which receives the output of the shaper 33 as an input signal and as a trigger signal and produces a sinusoidal output signal at the selected harmonic $2f_0$. This sinusoidal signal is sampled with an analog to digital converter 35 to produce a reference sequence $x(n)$ of digitised samples which are supplied as data to a processor and memory unit 36.

Mounted within the motor car passenger compartment are two loudspeakers 37_1 , and 37_2 , which are in positions normally used for car stereo reproduction. The loudspeakers are driven by a multiplexer 38 through respective low pass filters 39 and output amplifiers 40. The filters 39 have a cut off frequency of 460 Hz and are provided to prevent aliasing. The multiplexer 38 is controlled by the processor and memory unit 36, through control lines which are not shown, and receives a single input signal from a digital to analog converter 41. The purpose of the loudspeakers 37_1 and 37_2 is to generate in the passenger compartment audio waves that will cancel those set up directly by mechanical transmission from the engine to the compartment. The digital to analog converter 41 is supplied by the processor and memory unit 36 with output data which consists of two interleaved sequences of digitised samples $y_1(n)$ and $y_2(n)$ which are converted by the converter 41 into interleaved sequences of analog samples and separated into respective sequences by the multiplexer 38 for application to the low pass filters 39 so that in effect the loudspeaker 37_1 is driven by the sequence $y_1(n)$ and the loudspeaker 37_2 is driven by the sequence $y_2(n)$. In Fig. 5 each sequence is represented by the expression $y_m(n)$, so that in this example m may be 1 or 2.

In order to ensure that the acoustic outputs from the loudspeakers 37_1 and 37_2 have the correct phase and amplitude to effect cancellation of the engine noise, error signals are picked up from the compartment and utilised by the processor and memory unit 36. The acoustic error signals, if present, are sensed by four microphones 42_1 , 42_2 , 42_3 and 42_4 , which are placed respectively either side of a driver headrest and a passenger headrest, there being only two seats in the compartment in the present example. The electrical outputs from the microphones are respectively amplified by amplifiers 43 and passed through low pass filters 44 to a four-input multiplexer 45 which supplies a single analog output to an analog to digital converter 46. The filters 44 are provided to prevent aliasing and have a cut-off frequency of 460 Hz.

The multiplexer 45 and the converter 46 convert the four filtered microphone outputs into data stream consisting of four interleaved sequences of digitised samples $e_1(n)$, $e_2(n)$, $e_3(n)$, and $e_4(n)$, which correspond respectively to the filtered outputs of the microphones 42_1 , 42_2 , 42_3 and 42_4 . In Fig. 5, each sequence is represented by $e_l(n)$ so that in this example l may be 1, 2, 3 or 4.

The processor and memory suit 36 receives a square wave at 1.2 kilohertz from a sample rate oscillator 47 which determines the rate at which the converters 35, 41 and 46 convert samples and the frame duration of processing carried out by the unit 36. Thus in the present example, the unit 36 completes its processing frame within 833 milliseconds. A crystal clock oscillator (not shown) with a frequency of 10 Megahertz is included in the unit 36.

The unit 36 simulates two adaptive filters, each having two coefficients, so that:

$$y_m(n) = w_{m0}x(n) + w_{m1}x(n - 1)$$

describes the relation between the output sequence $y_m(n)$ to a loudspeaker and the reference signal $x(n)$, where the coefficients are w_{m0} and w_{m1} . Hence with the two loudspeakers 37_1 and 37_2 :

$$y_1(n) = w_{10}x(n) + w_{11}x(n - 1) \text{ and}$$

$$y_2(n) = w_{20}x(n) + w_{21}x(n - 1).$$

The values of the coefficients w_{m0} and w_{m1} are calculated by the unit 36 from the relationship:

$$w_{mi}(n+1) = w_{mi}(n) + \alpha \sum_{l=1}^4 e_l(n) r_{lm}(n-1)$$

in which α is a fixed convergence coefficient, $r_{lm}(n-1)$ is a value of a filtered reference signal r_{lm} , and $i = 0$ or 1 .

The filtered reference signal r_{lm} is a sequence formed by filtering the reference signal $x(n)$ with a filter that models the effect of the acoustic coupling between the m^{th} loudspeaker and the l^{th} microphone. The unit 36 simulates this filtering as digital FIR (Finite Impulse Response) filtering. Coefficients for the digital FIR filtering are adjusted adaptively during an initialisation program in which a white noise generator 48 is energised.

In the initialisation program, a white noise signal is generated by the generator 40, filtered by a low pass filter 49 to prevent aliasing, the filter having a cut off frequency of 460 Hz, and subsequently sampled and converted by an analog to digital converter 50. The digital output of the converter 50 is used to drive the loudspeakers 37_1 and 37_2 , and the resulting digital input to the unit 36 from the microphones 42_1 , 42_2 , 42_3 and 42_4 is used to determine the values of reference filter coefficients c_{mj} where $j = 0, \dots, 34$. The unit 36 performs a 35 coefficient FIR modelling of the impulse response between the m^{th} loudspeaker and the l^{th} microphone at the j^{th} sample. Such modelling is described in "Adaptive Signal Processing" by B. Widrow and S.D. Stearns, published in 1985 by Prentice Hall.

The filtered reference sequence is then given by

$$r_{lm}(n) = \sum_{j=0}^{34} c_{lmj} x(n-j)$$

The operation of the unit 36 is such that, having obtained the error samples $e_l(n)$ and the filtered reference signal $r_{lm}(n)$, each adaptive filter coefficient w_{mi} for each output $y_m(n)$ is updated by a quantity proportional to the sum of the computed products of $e_l(n)$ and $r_{lm}(n-1)$ in accordance with the equation

$$w_{mi}(n+1) = w_{mi}(n) + \alpha \sum_{l=1}^4 e_l(n) r_{lm}(n-1)$$

The new set of adaptive coefficients w_{mi} is then stored and used to filter the next sample of the reference signal, $x(n+1)$.

The unit 36 includes RAM for temporary storage and computation,

and EPROM for program storage. Calculated coefficients w_{mi} and c_{lmj} , and reference sequences $r_{lm}(n)$ are held in RAM. The convergence coefficient α is entered at a set of manually operable switches.

In a preferred embodiment constructed in accordance with Fig. 5, the unit 36 includes a Texas Instruments TMS 32010 microprocessor. The input signal rate from ignition circuit in the constructed embodiment is 100 Hz to 200 Hz, and the waveform shaper 33 is a monostable circuit triggered by the leading edge of the input signal to produce pulses of a constant width which is small relative to the sample period set by the sample rate of 1.2 kilohertz. The low pass filters 39, 44 and 49 are active filter modules supplied by Kemo Limited under no. 1431/L.

Only a small amount of separate additional RAM is required with the TMS 32010 which has considerable internal RAM and operates as described in the TMS 32010 Users' Guide published in 1983 by Texas Instruments, Inc. Data buses between the unit 36 and the converters 35, 41, 46 and 50 are 12 bit buses. Other buses and lines required for synchronisation and control are omitted for clarity. It will be noted that the values of the coefficients w_{mi} and c_{lmj} can be initially set to zero.

CLAIMS

1. An active noise control system for reducing sound or vibration in an enclosed space, generated by an exterior source, in which a reference signal, containing at least one selected harmonic or subharmonic of the sound or vibration generated by the external source, is supplied to means driving a plurality of secondary sources distributed in the enclosed space, such that the sound or vibration energy detected by sensor means distributed in the enclosed space is reduced.
2. A system according to claim 1, wherein the reference signal is generated by filtering a periodic input signal having its fundamental frequency locked to a predominant frequency of the exterior source.
3. A system according to claim 2, wherein the filtering is effected by a tracking filter.
4. A system according to claim 3, wherein the periodic input signal is a harmonic of the said predominant frequency.
5. A system according to any preceding claim, wherein the means for driving the secondary sources include a digital processor with data and program memory.
6. A system according to claim 1, wherein the reference signal is generated by apparatus as described hereinbefore with reference to Fig. 3 or Fig. 4 of the accompanying drawings.
7. An active noise control system substantially as described hereinbefore with reference to Fig. 5 of the accompanying drawings.

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